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# Corrective Filtration in the Processing of Electrical Signals to Improve Measurement Accuracy

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### Abstract

The research is devoted to the construction of an inverse correction filter, to isolate a useful signal that has been exposed to a direct filter. The structure of the reverse filter for each direct filter is individual and calculated based on the parameters of the direct filter. So, for a high-frequency finite difference filter, the filter of a finite sum is inverse. The orders of the forward and reverse filters must be coordinated. Using the software MATLAB, we assembled the circuit of the system, and investigated the effect of the reverse filter on the output signal of the system. The synthesis of the most optimal low-frequency filter having non-integer coefficients is carried out. In a cascade with a finite-difference filter, it suppresses all components of the error in the entire frequency band of the measuring signal.

### Keywords

Measuring System, Measurement Experiment, Measurement Information, Digital Signal Processing, Correcting Filtration, Reverse Filter, Filtering Algorithms, Adaptive Filters

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## 1. Introduction

The problem of interference in a signal containing also useful information about the observed process is being combated in various ways, including using filtering algorithms. The use of measuring devices (for example, motion sensors) coexists with the problem of interference in the signal. Analysis of real processes of movement of a well-known nature with an accessible physical model is accompanied by the need to receive indications without interference. They resort to signal filtering, which implies that information about signals and interference, for example, their statistical models (linearity of models, stationarity and normality) is available to some extent. However, in practice, statistical information is not always known. A special place among the algorithms intended for the identification of a useful signal is occupied by "adaptive filters - systems whose parameters adapt to a signal with a predetermined statistical model". In other words, the filters transform the system, changing its states depending on the statistical parameters. For example, the Kalman Filter is an adaptive filter that allows you to filter data using information about the physics of the process [5-7].

The research topic is relevant in the scientific field. The subject of the study is to reduce the amount of noise and noise by adapting the sensor readings, applying filters to improve the accuracy of the readings. Knowledge of the physical law of the observed process helps when choosing a filtering algorithm. The latter fact and the undeniable convenience of using filters make researchers resort to the use of filters to isolate a useful signal.

There are also other approaches to solving the problem of interference in the sensor readings. But, for example, by averaging the obtained indications, as proposed in the method of least squares, it will not be possible to analyze and evaluate the state of dynamical systems. Methods such as the least squares method can be applied when readings are

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available from the start point to the end of the reading. At the same time, all the accumulated reports are processed at once, and this can cause the processing time to increase, which is not always permissible. Of all the methods used for digital signal processing, the most important is digital filtering. In the past, interest in it has been limited to theoretical research, but recently it has been used in many important practical applications for processing the most complex signals, due to the availability of effective and relatively simple methods for constructing filters; success in the development of hardware and software for computers, especially with the advent of microprocessors and micro-computers, and the creation of fast matrix processors that are used as peripherals of computing systems or as the main processing system [1-3,].

As a result, digital filtering has been used recently in a variety of areas, and the task of isolating a useful signal from a mixture is an urgent task. Therefore, with respect to real measuring instruments, it is not optimal, but quasi-optimal (by some criteria) corrective filtering that is achievable. With this approach to filtering, taking into account the variety of structures, algorithms and practical implementation of the components of the measuring channel, it is advisable to orient certain methods of correcting the filtering to the forms of representation of the measuring signal (analog, digital, digital). Therefore, it is possible to distinguish three main varieties of corrective filtration depending on the subtract in which it is realized - continuous, discrete and digital.

## 2. Formulation of the Problem

Over the past decades, digital signal processing has become very important because now they not only replace classical analog methods in many traditional areas of technology but are also used in many new areas. Of all the methods used for digital signal processing, the most important is digital filtering. In the past, interest in it has been limited to theoretical research, but recently it has been used in many important practical applications for processing the most complex signals. This fact can be explained by the following reasons:

1. the availability of effective and relatively simple methods for constructing filters;
2. huge achievements in the field of high-integration microcircuit technology for multipliers, adders and memories with increased maximum operating frequency and in the development of new elements;
3. success in the development of hardware and software of computers, especially with the advent of microprocessors, and the creation of fast matrix processors that are used as peripherals of computer systems or as the main processing system.

As a result of these successes, digital filtering has recently been used in a wide variety of areas, such as radio communication, radio and sonar, physical experimentation, biomedical research, aerospace systems, mineral exploration using satellites, etc. Efficiency Corrective filtering, characterized by the degree of proximity of the measurement result to the true value of the informative parameter of the measuring signal, determines the achievable accuracy of the IR, which most objectively reflects the effectiveness of the entire digital measuring instrument.

When setting and solving problems, corrective filtering should take into account the following circumstances:

1. the input signal of each subtract can be represented as an additive mixture of useful signal and noise, the latter including systematic and random components;
2. a priori information about the useful signal and noise components of the error is incomplete because of their probabilistic-statistical nature and variability in the conditions of the functioning of the measuring channel;
3. due to the specifics of the tasks of transformation and processing of measurement information, the components of the measuring channel contain components with limited information, metrological, dynamic and operational characteristics;
4. elements of corrective filtration introduced into separate sub-contracts have different filtering properties and consequences, which can lead to the accumulation and transformation of error along the measuring channel;
5. most operators of digital processing of measuring information are linear, and therefore they are sensitive to both components of the processed additive mixture (signal + noise).

## 3. Resolving the Problem

The development of the theory of automatic control and the practice of creating automatic systems over the past decades is inseparably linked with the idea of using microprocessors in the processes of measuring and processing controlled parameters of objects [10-11].

The desire to design automatic systems that perform processing or measurement in the best way leads often to rather complex algorithms. It is clear that the result of these complex algorithms, as well as the imperfection of the technical means used, are the cause of the appearance of various kinds of errors and noise that can be represented as the difference between the measurement result and the true value of the measured quantity:

$$\varepsilon_i(t) = \theta_i^*\left[x(t)\right] - \theta\left[x(t)\right], \quad (1)$$

where, $\theta\left[x(t)\right]$ $u$ $\theta_i^*\left[x(t)\right]-$ respectively, the true and "noisy" values of the characteristic (parameter) of the process *x (t),* estimated under the measurement experiment.

Taking into account that most of the real errors are described by nonstationary ones in mathematical expectation, but stationary by dispersion and autocorrelation function by random processes, we present the model of the basic error of the measurement system in the following form:

$$\varepsilon(t) = \overline{\varepsilon}(t) + \varepsilon^0(t), \quad (2)$$

or:

$$\varepsilon(t) = \overline{\varepsilon}(t) + \varepsilon_l^0(t) + \varepsilon_h^0(t) + \varepsilon^0, \quad (3)$$

where $\overline{\varepsilon}(t)-$ the systematic component of the error, which is a non-stationary quantity (to it all the nonstationary error is attributed); $\varepsilon_l^0(t)$ $u$ $\varepsilon_h^0(t)-$ Low-frequency and high-frequency components, which are stationary centered random processes; $\varepsilon^0-$ a random centered value to which, for example, the analog-to-digital converter quantization noise, the computing noise of the computing devices.

Expression (1) defines the total (resulting) error, which consists of the components that depend on the properties of the measurement object, the signal *x(t)* and the measured characteristic, the measurement method, the algorithm implementation features, the measurement experiment, the object interaction, and the exchange of measurement information.

In the well-known scientific works and normative-technical documents various models of both the resultant error and its characteristic components are given.

In the partial classifications of errors, two approaches are distinguished. One approach is based on the choice as a classification feature, the factor causing the appearance of errors. This, for example, methodological and instrumental errors, errors due to the finite size of the sample, etc. Methodical errors are inherent in the accepted method of measurement. These include, for example, the quantization error in the level, the error due to the power consumption by the instrument from the source of the measured parameter, etc. The quantization error (noise) is typical for almost all digital measuring instruments, automatic control systems, communication systems, etc.

An enormous amount of work has been devoted to the study of the statistical properties and characteristics of quantization noise, and also to their influence on the measurement result. In these papers, the asymptotic properties of the process of quantization of continuous signals by level are determined and estimates of the parameters of the limiting pulse (digital) system using a quantizer with a sufficiently large number of quantization levels are obtained.

Under the second approach, the types of error, the nature of their manifestation in the measurement process, are taken for classification characteristics. This is a systematic and random error. The study of the components of error, correlated with the factors that determine them, allows us to identify the dominant causes of errors and identify ways to suppress them. In the study of errors, we subsequently use the second method of their classification. In rough measurements, it is assumed that the random error is independent and centered, and the systematic error is a constant or regularly changing. For a random component, this approach neglects that its characteristics can vary with time, and several error components can be stochastically interrelated.

With more stringent requirements for the accuracy of estimates of error values, it is necessary to take into account its frequency spectrum or autocorrelation function. Among the methods for reducing the correlated (systematic) component of the measurement error, the most effective are:

1. iterative methods;
2. method of exemplary measures;
3. test methods;
4. methods of auxiliary measurements;
5. method of negative feedback;
6. invariance method.

The measurement process is based on the following algorithm. In the first step, the measured value is connected to the input of the uncorrected measuring device by means of a distributor (switch). The noise present at the input and output of the uncorrected sub-tract of the exchange of measurement information is broadband, multicomponent and should be considered as nonstationary random processes (sequences). Therefore, the main requirement for corrective filters is the most complete suppression of filterable noise sequences for both components: non-stationary and centered stationary.

The corrective filter should be of a stopping nature. Therefore, it must include two filters in series: high-frequency (for suppressing low-frequency noise) and low-frequency (for suppressing high-frequency noise). Thus, with respect to suppression of all components of the resulting error of the measuring channel, the task of correcting filtering is significantly complicated. This dictates the need to develop universal methods to improve the accuracy of measurement results. The universality of the method of increasing accuracy is understood as its relevance to both the characters and the places of appearance and history of the components of the error of the measurement result.

Combating the accumulation of a progressive component of the error leads to the need for its localization and suppression (as far as possible) within the framework of single measurements. In the case of the presence of only a centered stationary error in the results of single measurements, a powerful tool for suppressing it is the use of the averaging method. However, the combination of these two procedures for suppressing errors within the framework of one general method of increasing accuracy is very problematic. Therefore, it is necessary to identify or determine a balanced series of procedures for localization and suppression of all components of error, which leads to the development of a universal method for increasing the accuracy of the results of digital dynamic measurements:

1. localization and suppression of progressive error within single measurements;
2. Suppression of the residual progressive error and the centered random stationary error at the stage of processing the results of single measurements.
3. The stage of processing the results of single measurements can be used:
4. when performing indirect measurements, for example, in flow metering;
5. with secondary corrective filtering of measurement data;
6. for statistical processing of useful and noise information in the process of metrological testing and attestation of the measuring device;
7. during the initial processing of the measuring information (linearization, scaling, etc.).

Proceeding from the above, we construct a measuring system with corrective filtration. At the input of the system (Figure 1), consisting of two filters - the finite difference filter and the inverse filter of the finite sum, a noisy harmonic signal is applied [9]. As a useful signal, a sinusoid is taken, white noise is the interference, and the amplitude of the useful signal is several times greater than the noise amplitude. At the input of the system, the useful signal and interference are added together in order to give a distorted signal to the input [4]. The distorted signal is fed to the input of the direct filter - the finite difference filter and to the input of the graphic display. Since the sinusoid is a low-frequency signal and the noise is high-frequency, the finite-difference filter therefore suppresses the low-frequency components of the signal, at the output of the direct filter we obtain pure noise, which is fed to the input of the reverse filter and to the input of the graphic display. At the output of the reverse filter, which restores the initial useful signal, we get a few purified harmonic signal, which is fed to the input of the graphic display. To evaluate the efficiency of the filter, it is necessary to calculate the filter suppression coefficient. For this, the variances at the input of the filter cascade and at its output are calculated, and their ratio is found [8, 12-14]. The more this ratio, the more effective the filter.

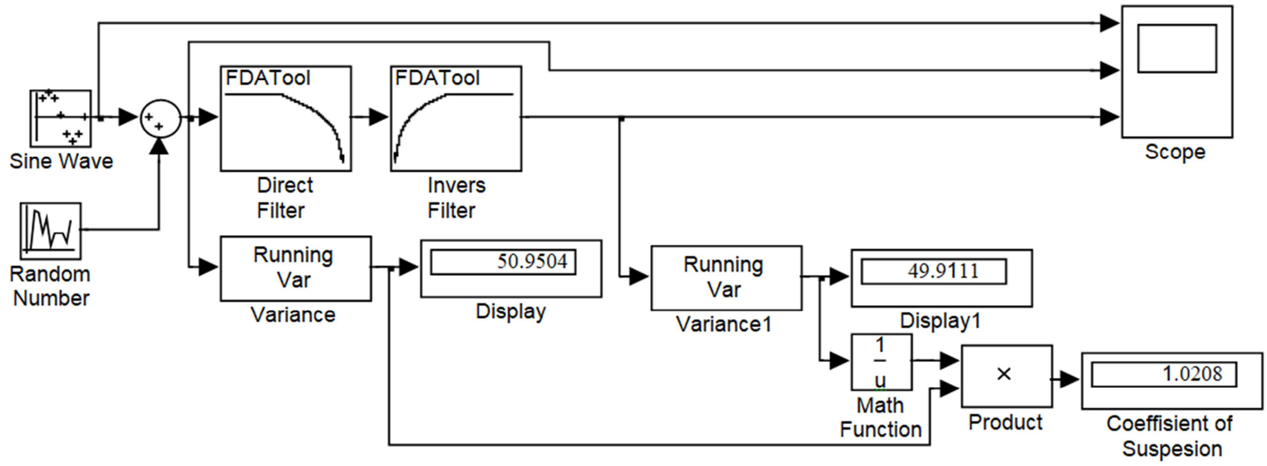


**Figure 1.** Measuring system with corrective filtration.

## 4. Conclusion

A review is made of the existing methods for increasing the accuracy of measuring dynamic quantities. It is proved that most of them affect only a certain component of the error- they suppress either an accidental or systematic error. It is proved that a filter that suppresses both the random and systematic components of the error is a filter of finite difference. It is also the most optimal among high-frequency filters for the criterion of minimum root-mean-square deviation of filter output noise. The synthesis of the most optimal low-frequency filter having non-integer coefficients is carried out. In a cascade with a finite-difference filter, it suppresses all components of the error in the entire frequency band of the measuring signal.

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